REMARKS/ARGUMENT

The foregoing Second Preliminary Amendment is being submitted to correct inadvertent typographical errors in the description of the prior art references contained in the specification. No new matter has been added.

Consideration and allowance of the application is earnestly solicited.

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Signature

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Date of Signature

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Respectfully submitted,

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APPENDIX B

VERSION WITH MARKINGS TO SHOW CHANGES MADE

37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

SPECIFICATION:

Paragraph at page 1, line 18:

With regard to CELP, a description is given in [M. Schroeder et al "Code excited linear prediction: High quality speech at very low bit rates" (Proc. ICASSP, pp. 937-940, 1985)] <u>M.R. Schroeder and Bishnu S. Atal "Code excited linear prediction (CELP): High quality speech at very low bit rates" (Proceeding of ICASSP, pp. 937-940, 1985)</u> (Reference 1). Further, a coding performance with regard to a music signal can be improved by constructing CELP, mentioned above, by a band division constitution. According to the constitution, a reproduction signal is generated by driving a linear prediction synthesis filter by an excitation signal provided by adding sound source signals in correspondence with respective bands.

Paragraph at page 2, line 1:

With regard to [CEIP] <u>CELP</u> having the band division constitution, a description is given in [A. Ubale et al "Multi-band CELP Coding of Speech and Music" (IEEE Workshop on Speech Coding for Telecommunications, pp. 101-102, 1997)] <u>A. Ubale and Allen Gersho "Multi-band CELP Coding of Speech and Music" (Proceeding of IEEE Workshop on Speech Coding for Telecommunications, pp. 101-102, 1997</u> (Reference 2).

Paragraph at page 26, line 1:

A linear prediction coefficient calculating circuit 170 inputs the input vector from the input terminal 10, carries out linear prediction analysis with regard to the input vector, calculates linear prediction coefficients αi, i=1, ..., Np, further, quantizes the linear prediction coefficients and calculates quantized linear prediction coefficients $\alpha i'$, i=1, ..., Np. Here, notation Np designates a linear prediction degree, for example, 16. Further, the linear prediction coefficient calculating circuit 170 outputs the linear prediction coefficients to a weighting filter 140 and outputs indexes in correspondence with the quantized linear prediction coefficients to a linear prediction synthesis filter 130, a linear prediction inverse filter 230 and a code outputting circuit 290. With regard to quantization of the linear prediction coefficient, there is, for example, a method of converting the linear prediction coefficient to a line

spectrum pair (LSP) and quantizing the converted linear prediction coefficient. With regard to conversion of the linear prediction coefficient into LSP, a description is given by [Sugamura et al "Speech information compression by a linear spectrum pair (LSP) speech analyzing and synthesizing system" (Proceeding of Electronic, Information and Communication Society A, Vol.J64-A, No. 8, pp. 599-606, 1981)] paragraph 3.2.3 of ITU-T Recommendation G.729, "Coding of Speech at 8 kbits/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACSLP)", 1996 (Reference 3). With regard to quantization of LSP, a description is given by [Omuro et al "Vector quantization of an LSP parameter by using moving average type interframe prediction" (Proceeding of Electronic, Information and Communication Society A, Vol.J77-A, No. 3, pp. 303-312, 1994) (Reference 4)] paragraph 3.2.4 of the Reference 3.

Paragraph at page 27, line 10:

Further, with regard to coding of a sound source signal, there can be used a method of efficiently expressing a sound source signal by a multiple pulse signal comprising a plurality of pulses and prescribed by positions of the pulses and amplitudes of the pulses. With regard to coding of a sound source signal using a multiple pulse signal, a description is given by [Ozawa et al "MP-CELP speech codification based on a multiple pulse spectra quantized sound source and high speed search" (Proceeding of Electronic, Information and Communication Society A, pp. 1655-1663, 1996)] paragraph 3.8.1 of the Reference 3, or paragraph 5.7 of GSM 06.60 version 6.0.1 Release 1997, "Digital Cellular Telecommunications System (Phase 2+); Enhanced Full Rate (EFR) Speech Transcoding" (ETSI EN 300 726, 2000) (Reference 4), or K. Ozawa and M. Serizawa, "High Quality Multi-Pulse Based CELP Speech Coding at 6.4 kbit/s and Its Subjective Evaluation" (Proceeding of ICASSP, pp. 153-156, 1998) (Reference 5). By the above described, an explanation of the first sound source generating circuit 110 is finished.

Paragraph at page 41, line 23 to page 42, line 3:

The down-sampling circuit 780 receives an input vector from the input terminal 10 and outputs a second input vector provided by down-sampling the input vector and having a first band to the first linear prediction coefficient calculating circuit 770 and the third differencer 183. Here, the first band is set to Fs1 [Hz] through Fe1 [Hz] similar to the first embodiment and a band of the input vector is set to Fs0 [Hz] through Fe0 [Hz] (third band). With regard to a constitution of the down-sampling circuit, a description is given to [paragraph 4.1.1 of a Reference (Reference 6) titled as "Multirate Systems and Filter Banks" by P.P. Vaidyanathan] paragraph 2.3.2 of a document titled as "Multirate Digital Signal

Processing" (Prentice-Hall Signal Processing Series, 1983) by R.E. Crochiere and L.R. Rabiner (Reference 6).

Paragraph at page 43, line 6:

The up-sampling circuit 781 receives the first reproduced vector outputted from the first linear prediction synthesis filter 132, upsamples the first reproduced vector and generates a third reproduced vector having a third band. In this case, the third band falls in a range of Fs0 [Hz] through Fe0 [Hz]. The up-sampling circuit 781 outputs the third reproduced vector to the first differencer 180. With regard to a constitution of the up-sampling circuit, a description is given to [paragraph 4.1.1 of the reference (Reference 6) titled as "Multirate systems and Filter Banks" by P.P. Vaidyanathan] paragraph 2.3.2 of a document titled as "Multirate Digital Signal Processing" (Prentice-Hall Signal Processing Series, 1983) by R.E. Crochiere and L.R. Rabiner (Reference 6).

Paragraph at page 43, line 26 to page 44, line 21:

The third linear prediction coefficient calculating circuit 772 is provided with a table stored with first quantized linear prediction coefficients. The third linear prediction coefficient calculating circuit 772 is inputted with the second linear prediction coefficient outputted from the second linear prediction coefficient calculating circuit 771 and the index in correspondence with the first quantized linear prediction coefficient outputted from the first linear prediction coefficient calculating circuit 770. The third linear prediction coefficient calculating circuit 772 reads a first quantized linear prediction coefficient in correspondence with the index from the table, converts the first quantized linear prediction coefficient into LSP, further and subjects LSP to sampling frequency conversion to thereby form first LSP in correspondence with a sampling frequency of the input signal. Further, the third linear prediction coefficient calculating circuit 772 converts the second linear prediction coefficient into LSP and generates a second LSP. The third linear prediction coefficient calculating circuit 772 calculates a difference between second LSP and first LSP. A difference value thereof is defined as third LSP. Here, with regard to the sampling frequency conversion of LSP, a description is given to [Japanese Patent Application No. 202475/1997 (Reference 7)] Japanese Unexamined Patent Publication (JP-A) No. 030997/1999 (Reference 7). The third LSP is quantized and the quantized third LSP is converted into a linear prediction coefficient and a third quantized linear prediction coefficient having the third band is generated. Further, the index in correspondence with the third quantized linear prediction coefficient is outputted to the linear prediction inverse filter 730 and the code outputting circuit 790.